Notice of Allowability	Application No.	Applicant(s)
	09/659,650	HAGIRAHIM ET AL.
	Examiner	**Art Unit**
	Blanche Wong	2616

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address--

All claims being allowable, PROSECUTION ON THE MERITS IS (OR REMAINS) CLOSED in this application. If not included herewith (or previously mailed), a Notice of Allowance (PTOL-85) or other appropriate communication will be mailed in due course. **THIS NOTICE OF ALLOWABILITY IS NOT A GRANT OF PATENT RIGHTS.** This application is subject to withdrawal from issue at the initiative of the Office or upon petition by the applicant. See 37 CFR 1.313 and MPEP 1308.

1. ☒ This communication is responsive to *Response under 37 C.F.R. 1.111 dated August 9, 2007*.
2. ☒ The allowed claim(s) is/are *1,6-13,18-27,32,33 (renumbered 1-21)*.
3. ☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
   a) ☐ All b) ☐ Some* c) ☐ None of the:
   1. ☐ Certified copies of the priority documents have been received.
   2. ☐ Certified copies of the priority documents have been received in Application No. _____ .
   3. ☐ Copies of the certified copies of the priority documents have been received in this national stage application from the International Bureau (PCT Rule 17.2(a)).

   * Certified copies not received: _____.

Applicant has THREE MONTHS FROM THE "MAILING DATE" of this communication to file a reply complying with the requirements noted below. Failure to timely comply will result in ABANDONMENT of this application.
THIS THREE-MONTH PERIOD IS NOT EXTENDABLE.

4. ☐ A SUBSTITUTE OATH OR DECLARATION must be submitted. Note the attached EXAMINER'S AMENDMENT or NOTICE OF INFORMAL PATENT APPLICATION (PTO-152) which gives reason(s) why the oath or declaration is deficient.
5. ☒ CORRECTED DRAWINGS (as "replacement sheets") must be submitted.
   (a) ☐ including changes required by the Notice of Draftsperson's Patent Drawing Review (PTO-948) attached
   1) ☐ hereto or 2) ☐ to Paper No./Mail Date _____.
   (b) ☒ including changes required by the attached Examiner's Amendment / Comment or in the Office action of Paper No./Mail Date Sept07.

   Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the drawings in the front (not the back) of each sheet. Replacement sheet(s) should be labeled as such in the header according to 37 CFR 1.121(d).
6. ☐ DEPOSIT OF and/or INFORMATION about the deposit of BIOLOGICAL MATERIAL must be submitted. Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Attachment(s)

1. ☐ Notice of References Cited (PTO-892)
2. ☐ Notice of Draftperson's Patent Drawing Review (PTO-948)
3. ☒ Information Disclosure Statements (PTO/SB/08), Paper No./Mail Date _____
4. ☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material
5. ☐ Notice of Informal Patent Application
6. ☐ Interview Summary (PTO-413), Paper No./Mail Date _____ .
7. ☒ Examiner's Amendment/Comment
8. ☒ Examiner's Statement of Reasons for Allowance
9. ☐ Other _____.

EXAMINER'S AMENDMENT

1. An examiner's amendment to the record appears below. Should the changes and/or additions be unacceptable to applicant, an amendment may be filed as provided by 37 CFR 1.312. To ensure consideration of such an amendment, it MUST be submitted no later than the payment of the issue fee.

Authorization for this examiner's amendment was given in a telephone interview with Michael Bentley on August 31, 2007.

The application has been amended as follows:

1. (currently amended) A method, comprising the steps of:

receiving [[a]] first voice traffic at a first Voice over Internet Protocol (VoIP) gateway;

determining whether a destination of the first voice traffic is serviced by a second VoIP gateway;

in response to a determination that said destination is serviced by said second VoIP gateway, multiplexing, at said first VoIP gateway, at least one modified RTP packet conveying said first voice traffic with at least one modified RTP packet conveying ~~said~~ second voice traffic ~~if said second voice traffic is~~ being provided to said second VoIP gateway; and

transporting said multiplexed ~~voice traffic~~ <u>modified RTP packets</u> to said second VoIP gateway utilizing a plurality of ~~transport packets, wherein said transport packets are~~ User Datagram Protocol (UDP)/Internet Protocol (IP) packets~~, and wherein said UDP/IP packets transport said modified RTP packets~~;

wherein <u>each of</u> said modified RTP packets ~~comprise~~ <u>comprises</u> ~~at least one of~~:

a Payload field ~~for containing a~~ <u>including</u> voice traffic;

<u>an RTP header;</u>

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying ~~the~~ <u>a</u> size of the ~~payload~~ <u>Payload</u> field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

2-5. (cancelled)

6. (previously presented) The method of claim 1, wherein said Header Error Check field performs one bit error correction.

7. (previously presented) The method of claim 1, further comprising the step of communicating messages between said first VoIP gateway and said second VoIP gateway.

8. (currently amended) The method of claim 7, wherein, during a call setup, said first VoIP gateway communicates an Open Logical Channel message to said second VoIP gateway, wherein said Open Logical Channel message includes ~~including~~ said first VoIP gateway's port number and a Call Identifier of the calling party.

9. (currently amended) The method of claim 8, wherein, in response to said Open Logical Channel message, said second VoIP gateway communicates an Open Logical Channel ACK message to said second VoIP gateway, wherein said Open Logical Channel ACK message includes ~~including~~ said second VoIP gateway's port number and a Call Identifier ~~for~~ of the called party.

10. (currently amended) The method of claim 7, wherein, in response to a caller terminating a call, said first VoIP gateway communicates a Close Logical Channel message to said second VoIP gateway, wherein said Close Logical Channel message includes ~~including~~ said first VoIP gateway's port number and ~~said~~ a Call Identifier of the calling party ~~to said second VoIP gateway~~.

11. (currently amended) The method of claim 10, wherein, in response to said Close Logical Channel message, said second VoIP gateway communicates a Close Logical Channel ACK

message to said first VoIP gateway, wherein said Close Logical Channel ACK message includes ~~including~~ said second VoIP gateway's port number and ~~said~~ a Call Identifier of the called party.

12. (original) The method of claim 1, wherein said step of determining is made utilizing a gatekeeper.

13. (currently amended) In a communication system for transporting voice traffic over an Internet Protocol (IP) network to a destination, apparatus comprising:

a first Voice over Internet Protocol (VoIP) gateway, for receiving [[a]] first voice traffic;

said first VoIP gateway determining whether said destination of said first voice traffic is serviced by a second VoIP gateway;

said first VoIP gateway, in response to a determination that said destination is serviced by said second VoIP gateway, multiplexing at least one modified RTP packet conveying said first voice traffic with at least one RTP packet conveying second voice traffic ~~if said second voice traffic is~~ being provided to said second VoIP gateway;

said first VoIP gateway transporting said multiplexed ~~voice traffic~~ modified RTP packets to said second VoIP gateway utilizing a plurality of ~~transport packets, wherein said transport packets are~~ User Datagram Protocol (UDP)/Internet Protocol (IP) packets~~, and wherein said UDP/IP packets transport said modified RTP packets~~;

wherein each of said modified RTP packets ~~comprise~~ comprises ~~at least one of~~:

a Payload field ~~for containing a~~ including voice traffic;

an RTP header;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying ~~the~~ a size of the ~~payload~~ Payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

14-17. (cancelled)

18. (previously presented) The apparatus of claim 13, wherein said Header Error Check field performs one bit error correction.

19. (previously presented) The apparatus of claim 18, further comprising the step of communicating messages between said first VoIP gateway and said second VoIP gateway.

20. (currently amended) The apparatus of claim 19, wherein, during a call setup, said first VoIP gateway communicates an Open Logical Channel message to said second VoIP gateway, wherein said Open Logical Channel message includes ~~including~~ said first VoIP gateway's port number and a Call Identifier of the calling party.

21. (currently amended) The apparatus of claim 20, wherein, in response to said Open Logical Channel message, said second VoIP gateway communicates an Open Logical Channel ACK message to said first VoIP gateway, wherein said Open Logical Channel ACK message includes ~~including~~ said second VoIP gateway's port number and a Call Identifier for the called party.

22. (currently amended) The apparatus of claim 21, wherein, in response to a caller terminating a call, said first VoIP gateway communicates a Close Logical Channel message to said second VoIP gateway, wherein said Close Logical Channel message includes ~~including~~ said first VoIP gateway's port number and ~~said~~ a Call Identifier of the calling party ~~to said second VoIP gateway~~.

23. (currently amended) The apparatus of claim 22, wherein, in response to said Close Logical Channel message, said second VoIP gateway communicates a Close Logical Channel ACK message to said first VoIP gateway, wherein said Close Logical Channel ACK message includes ~~including~~ said second VoIP gateway's port number and ~~said~~ a Call Identifier of the called party.

24. (original) The apparatus of claim 13, wherein a gatekeeper is used to determine whether said second VoIP gatekeeper services said destination.

25. (currently amended) A first Voice over Internet Protocol (VoIP) gateway for transporting voice traffic over an Internet Protocol (IP) network to a destination, comprising:

a processor; and

a storage device coupled to said processor and including instructions for controlling said processor, said processor operative with said instructions to:

receive [[a]] first voice traffic at said first VoIP gateway;

determine whether said destination of said first voice traffic is serviced by a second VoIP gateway;

in response to a determination that said destination is serviced by said second VoIP gateway, multiplex, at said first VoIP gateway, at least one modified RTP packet conveying said first voice traffic with at least one modified RTP packet conveying second voice traffic ~~if said second voice traffic is~~ being provided to said second VoIP gateway; and

transport said multiplexed ~~voice traffic~~ modified RTP packets to said second VoIP gateway utilizing a plurality of ~~transport packets, wherein said transport packets are~~ User Datagram Protocol (UDP)/Internet Protocol (IP) packets~~, and wherein said UDP/IP packets transport said modified RTP packets~~;

wherein each of said modified RTP packets ~~comprise~~ comprises ~~at least one of~~:

a Payload field ~~for containing~~ including a voice traffic;

an RTP header;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying ~~the~~ a size of the ~~payload~~ Payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

26. (currently amended) A first Voice over Internet Protocol (VoIP) gateway for transporting voice traffic over an Internet Protocol (IP) network to a destination as in claim 25, wherein a gatekeeper is used to determine whether said destination of said first voice traffic is serviced by said second VoIP gateway.

27. (currently amended) A first Voice over Internet Protocol (VoIP) gateway, for transporting voice over an Internet Protocol (IP) network, to a destination, comprising:

means for receiving [[a]] first voice traffic at said first VoIP gateway;

means for determining whether said destination of said first voice traffic is serviced by a second VoIP gateway;

means for multiplexing, at said first VoIP gateway, in response to a determination that said destination is serviced by said second VoIP gateway, at least one modified RTP packet ~~of~~ conveying said first voice traffic with at least one modified RTP packet ~~of~~ conveying second voice traffic ~~if said second voice traffic is~~ being provided to said second VoIP gateway; and

means for transporting said multiplexed ~~voice traffic~~ modified RTP packets to said second VoIP gateway utilizing a plurality of ~~transport packets, wherein said transport packets are~~ User Datagram Protocol (UDP)/Internet Protocol (IP) packets~~, and wherein said UDP/IP packets transport said modified RTP packets~~;

wherein each of said modified RTP packets ~~comprise~~ comprises ~~at least one of~~:

a Payload field ~~for containing a~~ including voice traffic;

an RTP header;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying ~~the~~ a size of the ~~payload~~ Payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

28-31. (cancelled)

32. (previously presented) The VoIP gateway of claim 27, wherein said Header Error Check field performs one bit error correction.

33. (currently amended) A method, comprising the steps of:

receiving [[a]] first voice traffic at a first Voice over Internet Protocol (VoIP) gateway;

transporting the first voice traffic to a second VoIP gateway utilizing a plurality of transport packets if a destination of the first voice traffic is serviced by the second VoIP gateway and second voice traffic is currently being provided to the second VoIP gateway, wherein the transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein each of said UDP/IP packets ~~transport~~ transports at least one modified Real-time Transport

Packet (RTP) packet ~~for each~~ of said first voice traffic and at least one modified RTP packet of said second voice traffic;

wherein each of said modified RTP packets ~~comprise~~ comprises ~~at least one of~~:

a Payload field ~~for containing a~~ including voice traffic;

an RTP header;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying ~~the~~ a size of the ~~payload~~ Payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

34. (cancelled)

2. The following changes to the drawings have been approved by the examiner and agreed upon by applicant: adding a Payload field in Fig. 4. In order to avoid abandonment of the application, applicant must make these above agreed upon drawing changes.

3. The following is an examiner's statement of reasons for allowance:

With regard to claims 1,13,25,27,33, the prior art of record fails to anticipate or make obvious "each of said modified RTP packets comprises: a Payload field including voice traffic, an RTP header, a Call Identifier field for identifying a caller; a Length Indicator field for identifying the size of the payload field; and a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field."

Any comments considered necessary by applicant must be submitted no later than the payment of the issue fee and, to avoid processing delays, should preferably

accompany the issue fee. Such submissions should be clearly labeled "Comments on Statement of Reasons for Allowance."

4. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Blanche Wong whose telephone number is 571-272-3177. The examiner can normally be reached on Monday through Friday, 830am to 530pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Edan Orgad can be reached on 571-272-7884. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

BW
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August 31, 2007

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